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Applicant's or agent's file reference AP582PCT	**IMPORTANT NOTICE**	
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Applicant BELL CANADA et al		

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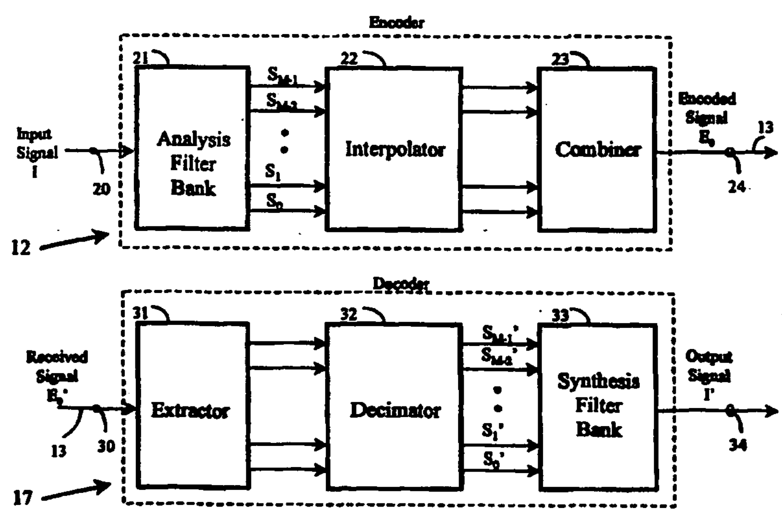

(57) Abstract

Improved transmission or storage of signals, such as high speed transmissions in subscriber loops of telecommunication systems, is facilitated by apparatus which includes an encoder (12) for encoding the input signal (I) before application to the transmission/storage medium (13) and a decoder (17) which decodes the signal received from the medium. The encoder comprises an analysis filter bank (21) which analyzes the input signal (I) into a plurality of subband signals (S_0, S_1, ...), each subband centered at a respective one of a corresponding plurality of frequencies; and a device (22) for combining a plurality of the interpolated subband signals to provide an encoded signal in which the interpolated subband signals occupy the same frequency band. The interpolated subband signals may be combined using QAM, CAP or DMT. The decoder (17) extracts the interpolated subband signals from a received encoded signal; and applies them to a synthesis filter bank, complementary and substantially inverse to the analysis filter, which processes the extracted interpolated subband signals (I') to produce a decoded signal corresponding to the input signal.

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DIGITAL TRANSMISSION USING SUBBAND CODING

DESCRIPTION

TECHNICAL FIELD:

(1) The invention relates to a method and apparatus for encoding digital signals for transmission and/or storage. The invention is especially, but not exclusively, applicable to the encoding of digital signals for transmission via communications channels, such as twisted wire pair subscriber loops in telecommunications systems, or to storage of signals in or on a storage medium, such as video signal recordings, audio recordings, data storage in computer systems, and so on.

BACKGROUND ART:

Using quadrature amplitude modulation (QAM), it is possible to meet the requirements for Asymmetric Digital Subscriber Loops (ADSL), involving rates as high as 1.5 megabits per second for loops up to 3 kilometres long with specified error rates or, 8 megabits per second over 1 kilometre loops. However, these rates are considered to be too low for proposed Very High Speed Digital Subscriber Loops (VDSL) which will require bit rates of, perhaps, 20 Mbits/second or higher. These rates cannot readily be achieved with conventional QAM systems in which there is a tendency, when transmitting at high bit rates, to lose a portion of the signal, particularly the higher frequency part, causing the signal quality to suffer significantly. This is particularly acute in two-wire subscriber loops, such as so-called twisted wire pairs.

It has been proposed, therefore, to use frequency division modulation (FDM) to divide the transmission system into a set of frequency-indexed sub-channels. The input data is partitioned into temporal blocks, each of which is independently modulated and transmitted in a respective one of the sub-channels. One such system, known as discrete multi-tone transmission (DMT), is disclosed in United States patent specification No. 5,479,447 issued December 1995 and in an article entitled "Performance Evaluation of a Fast Computation Algorithm for the DMT in High-Speed Subscriber Loop", IEEE Journal on Selected Areas in Communications, Vol. 13, No. 9, December 1995 by I. Lee *et al.*

Such DMT systems are not entirely satisfactory because noise and other sources of degradation could result in one or more sub-channels being lost. If only one sub-channel fails, the total signal is corrupted and either lost or, if error detection is employed, must be retransmitted. It has been proposed to remedy this problem by adaptively eliminating sub-channels, but to do so would involve very complex circuitry.

In United States patent specification No. 5,497,398 issued March 1996, M.A. Tzannes and M.C. Tzannes proposed ameliorating the problem of degradation due to sub-channel loss, and obtaining superior burst noise immunity, by replacing the Fast Fourier Transform with a lapped transform, thereby increasing the difference between the main lobe and side lobes of the filter response in each sub-channel. The lapped transform may comprise wavelets, as disclosed by M.A. Tzannes, M.C. Tzannes and H.L. Resnikoff in an article "The DWMT: A Multicarrier Transceiver for ADSL using *M*-band Wavelets", ANSI Standard Committee T1E1.4 Contribution 93-067, Mar. 1993 and by S.D. Sandberg, M.A. Tzannes in an article "Overlapped Discrete Multitone Modulation for High Speed Copper Wire Communications", IEEE Journal on Selected Areas in Comm., Vol. 13, No. 9, pp. 1571-1585, Dec. 1995, such systems being referred to as "Discrete Wavelet Multitone (DWMT).

A disadvantage of both DMT and DWMT systems is that they typically use a large number of sub-channels, for example 256 or 512, which leads to complex, costly equipment and equalization and synchronization difficulties. These difficulties are exacerbated if, to take advantage of the better characteristics of the two-wire subscriber loop at lower frequencies, the number of bits transmitted at the lower frequencies is increased and the number of bits transmitted at the higher frequencies reduced correspondingly.

The invention is particularly concerned with the use of an analysis filter bank to decompose the signal to be transmitted/stored into a plurality of subband signals. The term "subband signals" sometimes is used to refer to a plurality of narrowband signals within a prescribed band of interest, perhaps obtained by a corresponding plurality of bandpass filters. The present invention, however, is concerned with subband signals produced by an analysis filter bank, as disclosed in an article entitled "Perfect-channel Splitting By Use of Interpolation and Decimation Tree Decomposition Techniques", Proc. Intl. Conf. Inform. Sci. Syst., pp. 443-446, Aug. 1976, by A. Crosier, D. Esteban and C. Galand. Crosier *et al.* disclosed an analysis filter bank in the form of a

quadrature mirror filter (QMF) bank. For a more recent discussion of subband transforms, which include certain wavelet transforms, the reader is directed to an article entitled "Wavelet and Subband Transforms: Fundamentals and Communication Applications", Ali N. Akansu *et al.*, IEEE Communications Magazine, Vol. 35, No. 12, December 1997. Both of these articles are incorporated herein by reference.

As explained by Akansu *et al.*, in discrete-time signal processing, the sampling rate can be reduced by splitting the frequency band using Finite Impulse Response (FIR) filters. Each time one splits the band into a number of parts, for example halves, the sampling rate is reduced by the same number. Hence, repeatedly splitting the frequency band can enable the signal to be processed at a much lower sampling rate. Unfortunately, overlap between the different bands causes aliasing problems. As discussed by Crosier *et al.*, to split the frequency band exactly so as to allow "perfect reconstruction" of the original signal would require using so-called "brick" filters, i.e. FIR filters of infinite duration, which neither overlapped nor left gaps. In practice, this is not possible, so Crosier *et al.* proposed limiting the time duration using weighting functions which allow the subbands to overlap yet provide perfect reconstruction, at least theoretically. Difficulties arise in choosing a weighting function such that the reconstructed signal is distortion free or a so-called "perfect reconstruction". However, providing the analysis filter bank and synthesis filter bank satisfy certain conditions, as set out in the article by Akansu *et al.*, "perfect reconstruction" can be achieved.

In a practical implementation, such as in a telecommunications system, some distortion may be acceptable, so it may be possible to use an analysis filter bank which does not quite meet the conditions set out in Akansu *et al.*'s article, and provides only so-called "pseudo perfect reconstruction".

In the context of the present invention, and hereafter in this specification, the term "analysis filter bank" refers to a filter bank meeting the afore-mentioned conditions for "perfect reconstruction", or the conditions for "pseudo-prefect reconstruction", and the term "subband signals" refers to signals produced by such an analysis filter bank.

In the analysis filter bank, the original signal is divided into a plurality of narrowband signals, each having a bandwidth considerably narrower than the bandwidth of either the original signal or the transmission channel through which the signal is to be transmitted. However, each of these narrowband signals must be downsampled to create a subband signal. The downsampling produces a subband signal which has a

frequency spectrum having a bandwidth much greater than not only the original narrowband signal but also the transmission medium.

Practical applications of such subband signals have been disclosed, for example, in US Patent No. 5,214,678, in which J.D. Rault disclosed a system which uses an analysis filter bank to divide a digital audio signal into a plurality of subbands, and in U.S. Patent No. 5,161,210 in which Druydesteyn disclosed a system in which an analysis filter bank splits an audio signal into subband signals and an auxiliary tone is injected into each of the subband signals which then are recombined using a synthesis filter.

It was generally accepted, therefore, that the subband signals could not be combined (without reconstruction) before transmission/storage.

In International patent application number WO 9809383, the present inventor Yeap *et al.* disclosed a technique for combining subband signals before transmission and/or storage which uses an analysis filter bank to produce a plurality of subband signals, interpolates the subband signals and combines the interpolated signals for transmission/storage, specifically by using them to modulate a corresponding plurality of carrier signals which then are added to form the encoded signal to be transmitted and/or stored. Upon reception/retrieval, the signal is decomposed into subband signals again which are decimated and applied to a synthesis filter to reconstruct the original signal.

Although interpolating the subband signals, and using them to modulate one or more carrier signals to be transmitted as a single signal, avoids duplication of transmission/recording channels and allows better recovery of the original signal in the presence of attenuation and radio frequency interference, i.e. from radio stations, the system disclosed in WO 9809383 is not entirely satisfactory. One disadvantage is that the transmitted/recorded signals may still be susceptible to impulse noise, which characteristically occupies a very wide frequency spectrum for a relatively short duration. Another disadvantage is that the modulation of carriers results in a passband signal only, and the bandwidth is proportional to the number of carriers, which have different center frequencies.

The present invention seeks to mitigate one or more of the disadvantages of the known systems and provide a method and apparatus for encoding digital signals using

subband signals, specific embodiments of which are less susceptible to distortion and others to corruption by impulse noise.

DISCLOSURE OF INVENTION:

According to one aspect of the present invention, there is provided apparatus comprising an encoder for encoding a digital input signal for transmission or storage and a decoder for decoding such encoded signal to reconstruct the input signal, the encoder comprising analysis filter bank means for analyzing the input signal into a plurality of subband signals, each subband centered at a respective one of a corresponding plurality of frequencies, interpolation means for upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals all occupying the same frequency band; and combining means for combining the interpolated subband signals to form the encoded signal for transmission or storage; and the decoder comprising synthesis filter bank means complementary to said analysis filter bank means for producing a decoded signal corresponding to the input signal, extracting means for extracting the interpolated subband signals from the received encoded signal; and decimator means for decimating each of the plurality of extracted interpolated subband signals to remove an equivalent number of samples to those interpolated during encoding and applying the decimated signals to the synthesis filter bank means, the synthesis filter bank means processing the plurality of decimated signals to reconstruct said input signal, characterized in that, in the encoder, the combining means so combines the plurality of interpolated subband signals within the encoded signal that said interpolated subband signals occupy the same frequency band and the decoder separates the interpolated subband signals extracted from the same frequency band.

In one embodiment of this first aspect of the invention, the combining means combines at least one pair of said interpolated subband signals substantially orthogonally to provide a combined signal comprising two orthogonal components each containing information from both subbands, and uses said combined signal to provide said encoded signal, and in the decoder, the means for extracting the interpolated subband signals comprises means for orthogonally demodulating the received signal to extract said pair of subband signals.

It should be noted that the term "substantially orthogonally" embraces signals which are orthogonal or pseudo-orthogonal.

Preferably, the combining means uses quadrature amplitude modulation or (QAM) or carrierless amplitude modulation (CAP) to provide such orthogonal modulation.

In a second embodiment of the invention, the combining means time-division multiplexes the plurality of interpolated subband signals to form said encoded signal and, in the decoder, the extracting means use time-division demultiplexing to extract the received interpolated subband signals.

Preferably the time-division multiplexing is performed before the upsampling and interpolation.

According to second and third aspects of the invention, there are provided the aforementioned encoder *per se* and aforementioned decoder *per se*.

According to a fourth aspect of the invention, there is provided a method of encoding a digital input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, comprising the steps of:

(i) using an analysis filter bank means, analyzing the input signal into a plurality of subband signals, each subband centered at a respective one of a corresponding plurality of frequencies;

(ii) upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals all occupying the same frequency band; and

(iii) combining the interpolated subband signals to form the encoded signal;

the decoding comprising the steps of:

(iv) extracting the interpolated subband signals from the received encoded signal;

(v) decimating each of the plurality of extracted interpolated subband signals to remove a number of samples equivalent to those interpolated during encoding; and

(vi) using synthesis filter bank means complementary to said analysis filter bank means, processing the plurality of decimated subband signals to reconstruct said input signal, characterized in that, during encoding, the plurality of interpolated subband signals are so combined within the encoded signal that said interpolated subband signals occupy the same frequency band and the decoder separates the interpolated subband signals during extraction.

In one embodiment of this fourth aspect of the invention, the combining of the interpolated subband signals combines at least one pair of said interpolated subband signals substantially orthogonally to provide a combined signal comprising two

orthogonal components each comprising information from both of the interpolated signals, and uses said combined signal to provide said encoded signal having two spectral lobes, each containing information from both interpolated subband signals, and at a decoder, the step of extracting the interpolated subband signals uses orthogonal carrier signals to demodulate the received signal and extract said pair of interpolated subband signals from the received encoded signal.

Preferably, the orthogonal combining of the interpolated subband signals uses QAM or CAP, and the extraction of the interpolated subband signals from the received signal uses complementary demodulation.

In a second embodiment of this fourth aspect of the invention, the plurality of interpolated subband signals are time-division multiplexed to form the encoded signal and, in the decoder, the received signal is time-division demultiplexed during extraction of the received plurality of interpolated subband signals.

Preferably the time division multiplexing is performed before the upsampling and interpolation, and the demultiplexing and downsampling conversely.

According to fifth and sixth aspects of the invention, there are provided the aforementioned encoding steps *per se* and aforementioned decoding steps *per se*.

The upsampling produces duplicates, and the interpolation filters select corresponding duplicates of the plurality of upsampled subband signals so that interpolated subband signals have substantially identical bandwidths which, preferably, optimize transmission/storage channel bandwidth utilization. For subscriber loops, a lowpass filter may be used for interpolation and duplicate selection. For other applications, a bandpass or highpass filter may be used.

In embodiments of any aspects of the invention, the upsampling rate may be equal to the number of subbands produced by the analysis filter bank means.

BRIEF DESCRIPTION OF THE DRAWINGS:

Various objects, features, aspects and advantages of the present invention will become more apparent from the following detailed description, of preferred embodiments of the invention, taken in conjunction with the accompanying drawings, which are described by way of example only.

Figure 1 is a simplified block schematic diagram illustrating a transmission system including an encoder and decoder embodying the invention;

Figure 2 is a block schematic diagram of the encoder and decoder of Figure 1;

Figure 3 is a detail diagram of an analysis filter bank and a synthesis filter bank of the encoder and decoder, respectively;

Figure 4 is a simplified block schematic diagram of an encoder which uses TDM to combine the interpolated signals;

Figure 5 is a TDM decoder complementary to the encoder of Figure 4;

Figure 6 is a block schematic diagram of an alternative TDM encoder using only two subband signals;

Figure 7 is a block schematic diagram of a TDM decoder complementary to the encoder of Figure 6;

Figure 8 illustrates a delay bank suitable for use in either the encoders or the decoders;

Figure 9A illustrates the frequency spectrum of an input signal I;

Figure 9B illustrates the frequency spectra of two narrowband signals produced in the analysis filter bank, before downsampling;

Figure 9C illustrates the spectra of the subband signals produced by downsampling the narrowband signals;

Figure 9D illustrates the spectra of the subband signals after upsampling;

Figure 10 illustrates the frequency spectrum of the encoded signal S′ from the TDM encoder of Figure 6;

Figure 11 is a block schematic diagram of an encoder which uses quadrature amplitude modulation (QAM) of two subband signals;

Figure 12 is a block schematic diagram of a QAM decoder for decoding signals from the encoder of Figure 11;

Figures 13A and 13B illustrate frequency spectra of the interpolated subband signals obtained by filtering the upsampled subband signals;

Figure 14 illustrates the frequency spectrum of the output signal of the QAM encoder;

Figure 15 illustrates, as an example, in the time domain, a much simplified two-tone input signal I applied to the QAM encoder of Figure 11;

Figure 16 illustrates the frequency spectrum of the input signal I;

Figures 17A, 17B, 17C and 17D illustrate the subband signals S_0, S_1, S_2 and S_3, respectively, produced by analysis filtering of the input signal I of Figure 16;

Figure 18 illustrates, in the time-domain, the encoded signal E_0 obtained by quadrature amplitude modulation of subband signals S_0 and S_1;

Figure 19 illustrates the frequency spectrum of the encoded signal E_0;

Figure 20 illustrates the corresponding decoded signal I′

Figure 21 illustrates, in the time domain, the encoded signal E_0'' following optional bandpass filtering;

Figure 22 illustrates the frequency spectrum of the encoded signal E_0'' following optional bandpass filtering;

Figure 23 illustrates the decoded signal I′ obtained by decoding the bandpass-filtered signal E_0'';

Figure 24 illustrates a multiresolution analysis filter bank which provides multi-resolution by means of three-stage Discrete Wavelet Transform decomposition using a pyramid algorithm; and

Figure 25 illustrates a corresponding multi-resolution synthesis filter bank using three-stage synthesis.

BEST MODES FOR CARRYING OUT THE INVENTION

A transmission system embodying the present invention is illustrated in Figure 1. The system comprises digital input signal source 10, a transmitter 11, an encoder 12, transmission medium 13, decoder 17, receiver 18 and signal destination 19. The transmitter 11 may use a known form of modulation, such as Quadrature Amplitude Modulation (QAM), Carrierless Amplitude/Phase Modulation (CAP), Discrete Multitone Modulation (DMT) or a variety of other forms of modulation that produce a digital signal I directly; or produce a signal that can be sampled to produce digital signal I. Encoder 12 encodes the digital signal I to provide an encoded signal E_0 which it applies to the transmission medium 13. The transmitter 11 and receiver 18 may be constructed in a manner that is well known to persons skilled in this art and so will not be described in detail here.

The encoder 12 and decoder 17 are shown in more detail in Figure 2. Input signal I from signal source 10 is applied to the encoder 12 in which it is encoded by means of an analysis filter bank 21 to provide subband signals. Interpolator 22 interpolates the subband signals so that the interpolated subband signals all occupy the same frequency band, and combining means 23 combines them and supplies the resulting

encoded signal E_0 to transmission medium 13, which is represented by a transmission channel 14, noise source 15 and summer 16, the latter combining noise with the signal in the transmission channel 14 before it reaches the decoder 17. The decoder 17 uses an extraction means 31 to separate the received interpolated subband signals and a decimator 32 to decimate the extracted interpolated subband signals at the same rate as the upsampling used in the encoder. Synthesis filter bank 33 processes the decimated signals to provide a decoded signal I′ which is a "perfect" or "pseudo-perfect" reconstruction of the input signal I.

It should be appreciated that the transmission medium 13 could be a storage medium instead, such as a video/audio recorder, computer storage device, and so on, and the same schematic representation would apply by analogy.

The output of the decoder 17 is supplied to the receiver 18. The usable bandwidth of channel 14 dictates the maximum allowable rate at which a signal can be transmitted over the channel.

A suitable uniform four-channel analysis filter bank 21 and a complementary synthesis filter bank 33 are illustrated in Figure 3. The analysis filter bank 21 comprises four narrowband filters, namely lowpass filter h_0, bandpass filters h_1 and h_2, and highpass filter h_3, with their inputs connected in common to receive the digital input signal I and filter it to produce four narrowband signals S_0^*, S_1^*, S_2^* and S_3^*, respectively, which are downsampled by downsamplers 50, 51, 52 and 53 to produce four corresponding subband signals S_0, S_1, S_2 and S_3, respectively, which are supplied to the interpolator 22 (Figure 2). Assuming that the input signal I has a sample rate of N, the subband signals each have a sample rate of N/4.

In the complementary synthesis filter bank 33, the extracted decimated signals S_0', S_1', S_2' and S_3' are upsampled by upsamplers 54, 55, 56 and 57, respectively, and supplied to a lowpass filter h_0', two bandpass filters h_1' and h_2', and a highpass filter h_3', respectively. The filtered signals are summed by summing device 58 to form the decoded output signal I′.

It should be noted that some of the subband signal pairs in Figure 2 may not need to be transmitted. For example, they may contain little transmission power as compared with other subband signals. When these subband signals are not transmitted, the synthesis filter bank 33 shown in Figure 3 will insert "zero" level signals in place of the missing subband signals. The reconstructed signal I′ would then be only a close

approximation to the original input signal I. Generally, the more subbands used, the better the approximation.

Embodiments of the invention, which use Time-Division Multiplexing (TDM) to combine the interpolated subband signals, will now be described with reference to Figures 4 to 11.

Referring to Figure 4, in a TDM encoder 12, the digital input signal I is applied via input port 20 to an analysis filter bank 21 which subdivides the signal into a plurality of subband signals, as will be described in more detail later. A selection of the subband signals S_0, S_1 ... S_{L-1} are applied to a corresponding plurality of delay banks DB_0 through DB_{L-1}, respectively, in time division multiplexing (TDM) unit 60. The time division multiplexing unit 60 multiplexes the selected subset of subband signals S_0 to S_{L-1} to produce a serial signal. This is necessary because the analysis filter bank 21 generates the individual values of the subband signals S_0 to S_{L-1} simultaneously, i.e., in parallel.

The inputs of the delay banks DB_0 - DB_{L-1} are connected to the poles of a set of changeover switches C_0 - C_{L-1}, respectively. One terminal of each of the switches C_0 - C_{L-1} is connected to a corresponding one of the outputs of the analysis filter bank 21 and, with the exception of switch C_0, the other terminal is connected to the output of the preceding one of the delay banks DB_0 - DB_{L-1}. The other terminal of switch C_0 is connected to a suitable source of a zero value. Hence, a first setting of the switches C_0 - C_{L-1} connects the delay banks DB_1 through DB_{L-1} in parallel to respective outputs of the analysis filter bank 21 to receive the subband signals S_0 to S_{L-1}, respectively. A second setting of switches C_0 - C_{L-1} connects the delay banks DB_0 through DB_{L-1} in series with each other to receive a series of zero values. The switches C_0 - C_{L-1} are controlled by a transmit/transform control unit 61 which switches them alternately between their first and second settings. With the switches in the first setting, each of the delay banks DB_0 through DB_{L-1} is filled with a string of values of the corresponding subband signal, which values will be clocked into the delay bank under the control of a clock (not shown) until it reaches capacity. When the delay banks DB_0 through DB_{L-1} are all full, the control unit 61 operates switches C_0 - C_{L-1} and, in the second setting, the contents of the delay banks are clocked out serially.

In this particular embodiment, the computations by the analysis filter bank 21 are suspended while the TDM unit 60 is outputting the serial stream of values. As will be

described later, in practice, the delay banks DB_0 through DB_{L-1} could be duplicated so as to provide a continuous signal without interrupting computations.

The multiplexed values from the output of the time division multiplexing unit 60 are applied to an interpolation unit 62 which comprises an upsampler 63 and an interpolation filter 64. The upsampling rate P will be determined according to specific system requirements, such as the number of subbands (N) created by the analysis filter bank 21 and desired degree of expansion of each subband signal, but generally will be at such a rate that the bandwidth of each upsampled subband signal will be equal to or slightly less than the bandwidth of the transmission channel 14.

The interpolation filter 64 interpolates between each pair of actual sample values to provide an interpolated sample value for insertion in the or each intervening upsampled bit interval. The interpolated signal $S_i^{\#}$ and $S_j^{\#}$ are converted to an analog signal by a digital-to-analog converter 65, with a sampling rate F_0, and a lowpass filter 66 removes quantization noise before supplying the encoded signal E_0 to output port 24 connected to the transmission medium 13.

The processing of the received encoded signal E_0 in the decoder 17, to reconstruct the original signal I, will now be described with reference to Figure 5. In the decoder 17, the signal from the transmission medium 13 is received via port 30 and supplied to a typical receiver front end unit represented by amplifier 68 for amplifying the attenuated signal. The amplified signal from amplifier 67 is converted by an analog-to-digital converter 68 which has a sampling rate F_0 that is the same as the sampling rate of D/A converter 65 (Figure 4). Typically, if the upsampling rate P is equal to the downsampling rate, then output sampling rate F_0 is equal to L (number of selected subbands) multiplied by F_s, the input signal sampling frequency. Thus, for two subbands, $F_0 = 2F_s$. A bandpass filter 69 with a bandwidth equivalent to that of the transmission medium 13 filters the digital signal to remove noise occurring outside the channel bandwidth, following which the filtered digital signal is downsampled by a downsampler 70. The downsampling rate P is the same as the upsampling rate of upsampler 63 (Figure 4). The downsampled signal is demultiplexed by demultiplexing unit 71 which is similar in configuration to the multiplexer unit 60 of Figure 4 in that it comprises a series of delay banks DB_0, DB_1, ... DB_{L-1} interconnected by switches C_0 - C_{L-1} controlled by a transform/receive control unit 72, but differs in that they are connected in reverse order.

The transform/receive control unit 72 operates the switches C_0 - C_{L-1} so that the delay banks DB_0 through DB_{L-1} accumulate the values of the received signal and, when a complete sequence has been stored in the delay banks DB_0-DB_{L-1}, connects them in parallel to output their contents to the respective inputs of a synthesis filter bank 33 which recombines them to recover the original signal which it provides as output signal I′ at output port 34.

The analysis filter bank 21 and the synthesis filter bank 33 (Figure 3) are complementary and designed to provide "pseudo perfect reconstruction" as described earlier.

It is desirable for the analysis filter bank 21 to perform its computations continuously. An encoder which uses only two subband signals and processes them continuously will now be described, by way of example, with reference to Figure 16, in which components corresponding or identical to components in Figure 4 have the same reference numeral, but with a prime. Thus, the digital input signal I supplied to input port 20′ of analysis filter bank 21′ is applied in common to narrowband filters h_i and h_j. The narrowband signals from the filters h_i and h_j are downsampled by downsamplers 50′ and 51′, respectively, to form subband signals S_i and S_j. The two subband signals S_i and S_j are multiplexed by a time division multiplex unit 22′ which differs from that in Figure 4 in that it comprises two delay bank units 60A and 60B which both operate upon the two subband signals S_i and S_j alternately. There are only two subband signals, so each of the delay bank units 60A and 60B comprises two delay banks DB_0 and DB_1, which may be identical to those in Figure 4. The length of each of these delay banks DB_0 and DB_1 would be a design decision and typically might be 128 or 256 integers long. It should be noted that, in this case, each output from the analysis filter bank 21 (Figure 4) will be an integer number.

In delay bank unit 60A, switch C_O' connects the input of first delay DB_0' to either the output of the downsampler 50′ in analysis filter bank 21′ to receive subband signal S_i, or to a zero value point. Second switch C_1' connects the input of second delay bank DB_1' to the output of downsampler 51′ in the analysis filter 21′ to receive subband signal S_j; or to the output of delay bank DB_0'.

Delay bank unit 60B comprises two delay banks DB_2' and DB_3' and two switches C_2' and C_3' connected in a similar manner to those of delay bank unit 60A, but switches C_2' and C_3' are poled oppositely to switches C_0' and C_1' of delay bank unit 60A. The

outputs of delay bank units 60A and 60B are connected to respective terminals of a selector switch C_5, the pole of which is connected to the input of interpolating unit 62'.

The switches C_0' to C_5' are controlled by transmit/transform control unit 61' so that, when subband signal values are being clocked into the delay banks DB_0' and DB_1' of delay bank unit 60A, the values in the delay banks DB_2' and DB_3' of delay bank unit 60B are being clocked out to the interpolation unit 61' and replaced by zeros. Conversely, when the subband signal values are being clocked into delay banks DB_2' and DB_3', the values in delay banks DB_0' and DB_1' are being clocked out to the interpolation unit 61'.

As before, the interpolation unit 62' comprises an upsampler 63', which upsamples the time-multiplexed subband signals by a factor P which is related to the downsampling rate and the sampling rate F_s as described previously. The interpolation filter 64' may be a Raise-Cosine filter or other low pass filter, or a bandpass filter, depending upon the application. As previously described, the serial data stream then is converted by D/A converter 65' and the resulting analog signal filtered by lowpass filter 66' to remove quantization noise before being applied to the transmission medium 13 as signal E_0.

Referring now to Figure 7, the corresponding decoder 17' comprises an input port 30' whereby the received encoded signal is applied to an amplifier 67', A/D converter 68', filter 69' and downsampler 70' which process is in the manner previously described with reference to the corresponding components shown in Figure 7. Filter 69' will have the substantially same bandwidth as filter 64' in the encoder (Figure 6). Demultiplexing unit 71' is similar to the multiplexing unit 60' in the encoder of Figure 6 in that it comprises two delay bank units 71A' and 71B' connected in parallel between the output of downsampler 70' and synthesis filter bank 33'. Delay bank unit 71A' comprises two delay banks DB_0' and DB_1', and two changeover switches C_0' and C_1'. The output of delay bank DB_1' is connected to the pole of switch C_1', which has one of its terminals connected to one of the inputs of synthesis filter 33' and the other free. The output of delay bank DB_0' is connected to the pole switch C_0' which has one of its terminals connected to the input of delay bank DB_1' and the other terminal connected to the other input of synthesis filter 33'.

The second delay bank unit 71B' is similar to first delay bank unit 71A' in that it comprises delay banks DB_2' and DB_3' and switches C_2' and C_3'. The inputs of delay

bank DB_0' of delay bank unit 71A' and delay bank DB_2' of delay bank unit 71B' are connected to the first and second terminals, respectively, of a changeover switch C_5', the pole of which is connected to the output of downsampler 70'. The five switches C_0' - C_5' are controlled by transform/receive control unit 72'. The switches C_0' and C_1' are poled oppositely to switches C_2' and C_3' so that, when values of the downsampled received signal are being clocked into delay bank unit 71A, the previously-stored values are being clocked out of delay bank unit 71B' to the synthesis filter bank 33' and vice versa.

Each of the delay banks DB_0 - DB_{L-1} in Figures 4, 5, 6 and 7 is identical to the others. One of them is illustrated in Figure 8 and comprises a series of unit delay elements $Z^{-1} = M/F_s$ where M is the downsampling rate in the analysis filter bank 21 for a maximally decimated filter and Fs is the sampling frequency of the digital input signal I.

Operation of the encoder of Figure 6 will now be described with reference also to Figures 9A, 9B, 9C, 9D, 10A, 10B and 11 which illustrate, in very simplified form, the various signals in the encoder. Thus, Figure 9A shows the frequency spectrum of a digital input signal I having a bandwidth BW which would be about 8 MHz centered upon a centre frequency fc_1 of 5 MHz. These are typical values for a digital input signal I that is produced by Quadrature Amplitude Modulation (QAM) as proposed for Very high speed Digital Subscriber Loop (VDSL) applications. The maximum frequency of the input signal I is around 9 MHz which is somewhat less than half of the sampling frequency F_s used by the QAM system which, for example, might range from at least 24 MHz to as much as 50 or 60 MHz. The frequency spectrum is symmetrical about the zero ordinal, i.e., there are two conjugate lobes that are symmetrical about zero. The frequency spectrum of a digital input signal I produced by Carrierless Amplitude Phase Modulation (CAP) would be similar.

Figure 9B illustrates the two narrowband signals S_i* and S_j* which are obtained by passing the digital input signal I through bandpass filters h_i and h_j, respectively (Figure 6). At each side of the zero ordinal, the two narrowband signals S_i* and S_j* generally occupy higher- and lower- frequency portions of the bandwidth BW and overlap adjacent the centre frequency fc_1. This overlapping of the two narrowband signals S_i* and S_j* is characteristic of analysis filter banks used for so-called pseudo-perfect reconstruction or perfect reconstruction.

When the narrowband signals S_i* and S_j* are downsampled by a factor of M, their bandwidth increases by the factor M. For the example of Figure 9B, the subband signals S_i* and S_j* each occupy at least 6 MHz, so the downsampled signals S_i* and S_j* at the outputs of analysis filter bank 21′ (Figure 6) each occupy the available bandwidth for a sampling frequency F_s of about 24 MHz minimum, as illustrated in Figure 9C.

When the subband signals are upsampled by upsampler 63′ (Figure 6) the bandwidth of the frequency spectrum of each subband signal is reduced and a series of similar, i.e. duplicate, lobes are produced, each centered on a different frequency, as shown in Figure 9D. It should be appreciated that the duplicate lobes shown in Figure 9D would not necessarily be the same for other input signals.

Assuming that interpolation filter 64′ is a low frequency bandpass filter, it removes the higher frequency duplicates. As shown in Figure 10, which shows the frequency spectra of the two subband signals S_i' and S_j', after upsampling and filtering, the upsampling rate P is chosen so as to reduce the frequency spectrum so that it is approximately equal to that of the channel bandwidth BW, with centre frequency fc_2. For the QAM input signal, with F_s = 20 MHz and M equal to 4, the upsampling rate P would be equal to, or greater than M, i.e. 4. It should be noted that in Figure 10 two spectra are shown, one for S_i' and one for S_j'.

It should be noted that centre frequency fc_2 in Figure 10 is not necessarily the same as the centre frequency fc_1 (Figure 9A) of the input signal I. In the case of a QAM input signal I, fc_1 would be the carrier frequency. For other forms of modulation, it might be defined differently. The centre frequency fc_2 shown in Figure 10 can be controlled so that it is centred upon the bandwidth of the transmission channel 14 itself. This can be done by suitable selection of the characteristics of filter 64′ (Figure 6).

More particularly, filter 64′ could be a bandpass filter which would both remove lower frequency and higher frequency duplicate lobes, leaving an intermediate lobe.

It should be noted that the system will usually be designed so that the channel bandwidth BW_{CH} will be approximately the same as the bandwidth BW of the input signal I. However, the bandwidth BW of the input signal I will not extend to zero, whereas the bandwidth BW_{CH} of the channel conceivably could extend to zero. In the latter case, filter 64′ could be selected to produce a baseband output signal matching the channel bandwidth, thereby optimizing utilization of, for example, a twisted pair subscriber loop.

The filter 64′ filters both subband signals S_i and S_j, which alternate in the serial data stream leaving the TDM unit 60′. As can be seen from Figure 10, the frequency spectrum of transmitted subband signal S_i' (shown in broken lines) is almost, but not quite, identical to the frequency spectrum of subband signal S_j'. The two frequency spectra are shown superimposed because, while they occupy substantially much the same bandwidth, i.e. the same frequency band, they occur alternately in the time domain. Consequently, the signal bandwidth in the channel varies only slightly during the transmission as a result of fluctuations between S_i' and S_j'.

A person skilled in this art will be able to infer the corresponding operation of the decoder, and the signals therein, from the description of the encoder, so they will not be described here.

TAS B7 Whereas the encoder of Figure 6 upsamples the subband signals after they have been multiplexed, it should be noted that the individual subband signals could be upsampled and interpolated before being multiplexed. However, an advantage of multiplexing before upsampling is that fewer delay banks are needed. Operation of such a modified encoder would be analogous to that of the encoder of Figure 6 and so will not be described herein.

An embodiment of the invention using Quadrature Amplitude Modulation (QAM) rather than TDM will now be described with reference to Figures 11 and 12 in which components corresponding to those in Figures 2 - 7 have the same reference numerals but with a double prime.

In this embodiment, two higher subbands are generated but not transmitted. In the QAM encoder 12″ of Figure 11, the input signal I is supplied via input port 20 to an analysis filter bank 21″ which generates lowpass subband signal S_0, two bandpass subband signals S_1 and S_2, and the highpass subband signal S_3. In this implementation, only subband signals S_0 and S_1 are processed. Bandpass subband signal S_2 and highpass subband signal S_3 are discarded. Interpolator means 22″ interpolates each of the pair of subband signals S_0 and S_1, respectively, by the same factor P, where P is an integer, typically 8 to 24. Thus, within interpolator 22″, the subband signals S_0 and S_1 are upsampled by upsamplers 63_0″ and 63_1″, respectively, which insert zero value samples at intervals between actual samples. The upsampled signals then are filtered by two interpolation filters 64_0″ and 64_1″, respectively, which insert at each upsampled "zero" point a sample calculated from actual values of previous samples. The interpolation

filters $64_0''$ and $64_1''$ may be bandpass filters or lowpass filters, but lowpass filters, for example Raise-Cosine filters, are preferred so as to minimize intersymbol interference. The two interpolated subband signals $S^u{}_0$ and $S^u{}_1$ are supplied to quadrature amplitude modulator 74 which uses them to modulate in-phase and quadrature components f_I and f_Q of a carrier signal f_0, provided by an oscillator 76, the quadrature component being derived by means of a phase shifter 76. The modulator 74 comprises multipliers 77_0 and 77_1 which multiply the in-phase and quadrature components f_I, and f_Q by the two interpolated subband signals S_0, and S_1, respectively. The resulting two modulated carrier signals $S^m{}_0$ and $S^m{}_1$ are added together by a summer 78 to form the encoded signal E_0 for transmission by way of port 24″ to transmission medium 13″.

For reasons which will be explained later, the output of the summer 23 may be supplied to transmission medium 12 by way of a filter 79, as shown in broken lines. In this particular example, filter 79 is a bandpass filter.

At the corresponding QAM decoder 17″ shown in Figure 12, the signal E'_0 received at port 30″ is supplied to a QAM demodulator 81 which comprises multipliers 82_0 and 82_1, which multiply the signal E'_o by in-phase and quadrature components f_I and f_Q of a carrier signal f_0 from an oscillator 83, the quadrature signal f_Q being derived by way of phase shifter 84. The frequency f_o is the same as that used in the QAM encoder 12″. The resulting signals are passed through lowpass filters 85_0 and 85_1, respectively, which extract the upsampled versions $S^{u*}{}_0$ and $S^{u*}{}_1$ which then are decimated by decimators $70_0''$ and $70_1''$, respectively, of decimator 70″. The resulting recovered subband signals $S^*{}_0$ and $S^*{}_1$ are each supplied directly to a corresponding one of two inputs of a synthesis filter bank 33″ which uses them to construct the signal I′ which corresponds to a reconstruction of the input signal I. In the synthesis filter bank 33″ the highpass subband signals S_2 and S_3, which were not transmitted, are each replaced by a "zero" signal at the corresponding "higher" frequency inputs 86_2 and 86_3 of the synthesis filter bank 33″. The resulting output signal I′ from the synthesis filter bank 33″ is the decoder output signal supplied via output port 34″, and is a close approximation to the input signal I at the input to the encoder 21″ of Figure 11.

If the higher subband signals S_2 and S_3 were used, the encoder 12″ would duplicate interpolator 22″ and modulator 74 and the decoder 17″ would duplicate the DQAM 81 and decimator 70″ as appropriate and a suitable synthesis filter bank 33″ would use all four subband signals $S^*{}_0$, $S^*{}_1$, $S^*{}_2$ and $S^*{}_3$.

In the QAM encoder 12″, the various signals in the analysis filter bank 21″ and the interpolator 22″ will be the same as those shown in Figures 9A - 9D described previously, the interpolated signals S^u_0 and S^u_1 being shown again in Figures 13A and 13B.

As shown in Figure 14, following modulation by the QAM means 74, and providing the interpolation filters are bandpass filters, the output signal E_0 from the QAM means 74 has a spectrum which has two lobes, one each side of the center frequency f_0 of the carrier used by the QAM means 74. The center frequency of the lower frequency lobe is equal to $f_0 - \Delta$, and the center frequency of the upper frequency lobe is equal to $f_0 + \Delta$, where Δ preferably is equal to about one quarter of the bandwidth BW of the original input signal I. However, Δ may vary depending upon the complexity of the input signal and the design of the analysis filter bank 21″. The bandwidth BW′ is determined in dependence upon the sampling rate Fs of the digital input signal I.

In the aforementioned international patent application number WO 9809383, the subband signals were modulated onto separate carriers having different center frequencies so their frequency spectrum lobes were separated by a guard band and each lobe contained information from its own subband only. By contrast, in this QAM embodiment of the present invention, there is no need for a guard band between the lobes in the output signal E_0. (Figure 11). It should be noted that each lobe contains information from both of the subband signals S_0 and S_1. Thus, as illustrated in Figures 13 and 13, the information A contained in the input signal I is split into lower-frequency component L(A) in subband signal S_o and higher-frequency component H(A) in subband signal S_i. As shown in Figure 14, after quadrature amplitude modulation, each lobe of encoded signal E_0 contains some of both components L(A) and H(A). Consequently, if one lobe is corrupted, perhaps because of noise or attenuation of higher frequencies, it may still be possible to reconstruct the original signal I. Hence, the transmission is more robust.

It should be appreciated that, if signal compression is desired, perhaps because bandwidth is limited, one of the lobes need not be transmitted. If the lower lobe were to be discarded, a high pass filter could be used to filter the output from encoder 12″ in Figure 11. Conversely, if the higher lobe were to be discarded, a low pass filter could be used instead. In the QAM encoder shown in Figure 11, a bandpass filter 79 is shown

(in broken lines) for removing the higher lobe. Use of a bandpass filter rather than a low-pass filter allows the portions of the spectrum below and above the lower lobe to be used for other purposes.

It should be noted that this is not the same as single sideband transmission where, although each sideband contains the same information, it is derived from a single source via a single modulated carrier.

Examples of signals obtained by modelling the QAM encoder 12″ and decoder 17″ are illustrated in Figures 15 - 23.

Figure 15 illustrates, in the time domain, a very simple input signal I comprising two sinusoidal signals, of 400 Hz and 1200 Hz, respectively. Figure 16 illustrates the corresponding frequency spectrum of this two-tone input signal I.

Figures 17A, 17B, 17C and 17D illustrate the corresponding four subband signals S_0, S_1, S_2 and S_3, respectively, obtained by analysis filtering the input signal I. It should be noted that bandpass subband signal S_2 has little energy compared with signals S_0 and S_1 and the energy content of highpass subband signal S_3 is negligible. Hence subband signals S_2 and S_3 are not used in encoding the encoded signal E_0 which is illustrated in Figure 18. As shown in Figure 19, the frequency spectrum of the encoded signal E_0 comprises two lobes, with respective peaks at 1600 Hz and 2400 Hz, i.e. at an offset Δ of 400 Hz either side of a center frequency of 2000 Hz. Some bandpass filtering was applied to remove harmonics.

Figure 20 illustrates the corresponding decoded signal I′ and shows that the two tones of the original input signal I have been recovered, but without the portion corresponding to omitted subband signals S_2 and S_3. It will be appreciated that, with suitable equalisation, the original digital signal I could be recovered despite this portion of the signal not being transmitted.

As mentioned previously, a further reduction in bandwidth can be achieved by transmitting only one lobe of the encoded signal E_0, predicated upon the fact that each lobe contains information from both subbands. Thus, Figure 21 illustrates in the time domain the encoded signal E_0 after filtering by bandpass filter 79 to remove the higher-frequency lobe and Figure 22 illustrates the frequency spectrum of the filtered encoded signal E_0. Figure 23 illustrates the corresponding decoded signal I′ and shows that, despite the fact that one lobe was not transmitted, the two tones have been recovered by the decoder 17″.

It is envisaged that, instead of bandpass filter 79, other means could be used to eliminate one of the lobes of the encoded signal E_0 before transmission/storage. For example, filter 79 could be replaced by a Fast Fourier Transform device, a phase shifting and cancellation circuit, or other suitable means.

Embodiments of the invention which allow higher frequency components and lower frequency components to be intermixed and compressed into a narrower bandwidth than the original signal are especially useful for use with two-wire subscriber loops of telecommunications systems since such loops tend to attenuate higher frequencies disproportionately.

It should be noted that the subband signal bandwidths could be greater than one half of the original signal bandwidth BW, though still less than BW. This would allow a less expensive analysis filter bank to be used.

It should be appreciated that the quadrature amplitude modulation means 74 and interpolator 22″ could be combined in a Carrierless Amplitude/Phase (CAP) modulation means which would comprise an in-phase filter means and a quadrature filter means each of which integrates the interpolation of the corresponding subband signal with the multiplier function.

If more subband signal pairs were to be used, an interpolator 22″ and QAM 74 would be provided for each additional pair, which would also be interpolated at such a rate that all of the modulated carriers had the same bit rate.

For a large number of subbands, it might be preferable to use a uniform analysis filter bank. However, for many applications it is possible to use a multiresolution analysis filter bank instead, as will be described later.

Various modifications may be made to the above-described embodiments of the invention without departing from the scope and spirit of the present invention. For example, the analysis filter bank and synthesis filter bank could be like those described in WO 9809383, or other suitable filter banks such as those disclosed by Akansu.

The analysis filter bank 21 may be replaced by a multiresolution filter bank, for example an octave-band filter bank, perhaps implementing a Discrete Wavelet Transform (DWT) such as is disclosed in International patent application No. WO 9809383, to which the reader is directed for reference.

Such a three stage octave-band tree structure for Discrete Wavelet Transformation will now be described with reference to Figures 24 and 25, in which the same

components in the different stages have the same reference number but with the suffix letter of the stage.

Referring to Figure 24, the three decomposition stages A, B and C have different sampling rates. Each of the three stages A, B and C comprises a highpass filter 40 in series with a downsampler 41, and a lowpass filter 42 in series with a downsampler 43. The cut-off frequency of each lowpass filter 42 is substantially the same as the cut-off frequency of the associated highpass filter 40. In each stage, the cut-off frequency is equal to one quarter of the sampling rate for that stage.

The N samples of input signal I are supplied in common to the inputs of highpass filter 40A and lowpass filter 42A. The corresponding N high frequency samples from highpass filter 40A are downsampled by a factor of 2 by downsampler 41A and the resulting N/2 samples supplied to the output as the highpass wavelet S_3. The N low frequency samples from lowpass filter 42A are downsampled by a factor of 2 by downsampler 43A and the resulting N/2 samples supplied to stage B where the same procedure is repeated. In stage B, the N/2 higher frequency samples from highpass filter 40B are downsampled by downsampler 41B and the resulting N/4 samples supplied to the output as bandpass wavelet S_2. The other N/2 samples from lowpass filter 42B are downsampled by downsampler 43B and the resulting N/4 samples are supplied to the third stage C, in which highpass filter 40C and downsampler 41C process them in like manner to provide at the output N/8 samples as bandpass wavelet S_1. The other N/4 samples from lowpass filter 42C are downsampled by downsampler 43C to give N/8 samples and supplies them to the output as low-pass wavelet S_0.

It should be noted that, if the input signal segment comprises, for example, 1024 samples or data points, wavelets S_0 and S_1 comprise only 128 samples, wavelet S_2 comprises 256 samples and wavelet S_3 comprises 512 samples.

Instead of the octave-band structure of Figure 24, a set of one lowpass, two bandpass filters and one highpass filter could be used, in parallel, with different downsampling rates.

It should be appreciated that, where the analysis filter bank 21 implements DWT, the synthesis filter 33 will implement inverse DWT. The corresponding synthesis filter bank for implementing Inverse Discrete Wavelet Transform is shown in Figure 25 but will not be described in detail since its operation will be apparent to one skilled in this art.

In each of the above-described embodiments, it is not essential for upsampling rate P to be equal to the downsampling rate M used in the analysis filter bank 21. Using an upsampling rate P which is less than downsampling rate M could reduce the output sampling frequency F_0 and hence the complexity of the D-to-A converter.

An advantage of embodiments of the present invention is that the encoded signal may be passband or baseband, enabling selection of a particular transmission band according to the particular application.

It should be appreciated that the signal source 10 and the encoder 12 could be parts of a transmitter 11 having other signal processing circuitry. Likewise, the decoder 17 and signal destination 19 could be parts of a corresponding receiver 18.

INDUSTRIAL APPLICABILITY

It should be noted that the present invention is not limited to transmission systems but could be used for other purposes to maintain signal integrity despite noise and attenuation. For example, it might be used in recording of the signal on a compact disc or other storage medium. The storage medium can therefore be equated with the transmission medium 13 in Figure 1. It should be appreciated that the encoders and decoders described herein would probably be implemented by a suitably programmed digital signal processor or as a custom integrated circuit.

Embodiments of the present invention advantageously simplify the circuitry required to modulate the input digital signal and improve the tolerance of transmitted/stored signal to impulse noise.

CLAIMS:

Sub a1 1. Apparatus comprising an encoder (12) for encoding a digital input signal for transmission or storage and a decoder (17) for decoding such encoded signal to reconstruct the input signal, the encoder comprising analysis filter bank means (21) for analyzing the input signal into a plurality of subband signals, each subband centered at a respective one of a corresponding plurality of frequencies, interpolation means (22;62) for upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals all occupying the same frequency band; and combining means (23;60;74) for combining the interpolated subband signals to form the encoded signal for transmission or storage; and the decoder comprising synthesis filter bank means (33) complementary to said analysis filter bank means for producing a decoded signal corresponding to a reconstruction of the input signal, extraction means (31;71;81) for extracting the interpolated subband signals from the received encoded signal; and decimator means (32;70) for decimating each of the plurality of extracted interpolated subband signals to remove a number of samples corresponding to those interpolated during encoding and applying the decimated signals to the synthesis filter bank means (33), the synthesis filter bank means processing the plurality of decimated signals to reconstruct said input signal, **characterized in that**, in the encoder, the combining means (23;60;74) so combines the plurality of interpolated subband signals within the encoded signal that said interpolated subband signals occupy the same frequency band and the decoder separates the interpolated subband signals from within said same frequency band.

2. Apparatus as claimed in claim 1, **characterized in that** the combining means comprises modulation means (74) for using a pair of said interpolated subband signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase displaced by 90 degrees one relative to the other, and combining the modulated signals to provide said encoded signal, said encoded signal having two spectral lobes each comprising information from both of the subband signals;

and the decoder comprises demodulation means for orthogonally demodulating the subband signals extracted from the received encoded signal.

3. Apparatus as claimed in claim 2, **characterized in that** the modulation means (74) comprises quadrature amplitude modulation means for using each of the interpolated subband signals to modulate a respective one of an in-phase carrier signal and a quadrature carrier signal, the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other, and the demodulation means (81) comprises means for demodulating the received encoded signal using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

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4. Apparatus as claimed in claim 1, **characterized in that** the analysis filter bank (21) generates a plurality of pairs of subband signals and the combining means selects fewer than all of said pairs, the synthesis filter bank (33) compensating for the unused subband signals by substituting zero level signals.

5. Apparatus as claimed in claim 2 or 3, further **characterized by** means (79) for removing one of said spectral lobes from the encoded signal and providing the remaining one of said spectral lobes as said encoded signal.

6. Apparatus as claimed in claim 1, **characterized in that**, in the encoder, the combining means comprises multiplexing means for time division multiplexing (60) the interpolated subband signals to form said encoded signal, and the extracting means comprises demultiplexing means (71) for time-division demultiplexing the received signal to provide a plurality of received interpolated subband signals corresponding to said interpolated signals in the encoder.

7. Apparatus as claimed in claim 1, **characterized in that** the encoder comprises means (60) for time-division multiplexing the subband signals and the interpolation means (62) upsamples and interpolates the multiplexed subband signals to form the interpolated subband signals, and, in the decoder, the decimator means (70) downsamples the received signal and the decoder further comprises demultiplexing means (71) for time-division demultiplexing the downsampled received signal to extract the subband signals.

8. Apparatus as claimed in claim 6 or 7, **characterized in that** the upsampling means upsamples at a rate (P) that corresponds to the number of subbands (N) created by the analysis filter bank means.

9. An encoder as claimed in claim 6, 7 or 8, **characterized in that** the means (60) for time division multiplexing the subband signals comprises delay means (DB_0, DB_1, ...) for storing a series of values of each subband signal, the delay means being operable alternately between a first state wherein the delay means accepts values of the subband signals in parallel and a second state wherein the delay means outputs previously stored values serially.

10. Apparatus as claimed in claim 6, 7 or 8, **characterized in that** in the encoder, the means (60) for time division multiplexing the subband signals comprises first delay means (60A) and second delay means (60B) each for accepting a series of values of each subband signal, each delay means being operable alternately between a first state wherein the delay means accepts values of the subband signals in parallel and a second state wherein the delay means outputs previously stored values serially, the arrangement being such that, when the first delay means is in its first state accepting values of the subband signals, the second delay means is in its second state and outputting the subband signal values previously stored therein, and, in the decoder, the means (71) for time division demultiplexing the subband signals comprises first delay means (71A) and second delay means (71B) each for accepting a series of values of the received signal, each delay means being operable alternately between a first state wherein the delay means accepts values of the received signal serially and a second state where the delay means output previously stored values in parallel, the arrangement being such that, when the first delay means is in its first state accepting values of the received signal, the second delay means is in its second state and outputting the subband signal values previously stored therein.

11. An encoder (12) for encoding a digital input signal (I) for transmission or storage comprising analysis filter bank means (21) for analyzing the input signal (I) into a plurality of subband signals, each subband centered at a respective one of a corresponding plurality of frequencies, interpolation means (22) for upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals

all occupying the same frequency band; and combining means (23;60;74) for combining the interpolated subband signals to form the encoded signal for transmission or storage, **characterized in that** the combining means (23;60;74) so combines the plurality of subband signals within the encoded signal that said interpolated subband signals occupy the same frequency band.

12. An encoder as claimed in claim 11, **characterized in that** the combining means comprises modulation means (74) for using a pair of said interpolated subband signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase displaced by 90 degrees one relative to the other, and combining the modulated signals to provide said encoded signal, said encoded signal having two spectral lobes, each comprising information from both of the interpolated subband signals.

13. An encoder as claimed in claim 12, **characterized in that** the modulation means (74) comprises quadrature amplitude modulation means for using each of the interpolated subband signals to modulate a respective one of an in-phase carrier signal (f_I) and a quadrature carrier signal (f_Q), the in-phase carrier signal and the quadrature carrier signal having the same frequency (f_0) but phase-displaced by 90 degrees one relative to the other.

14. An encoder as claimed in claim 11, **characterized in that** the analysis filter bank means (21) generates a plurality of pairs of subband signals and the modulation means (74) modulates a selection of said pairs.

15. An encoder as claimed in any one of claims 11 to 14, further **characterized by** means (79) for removing one of said spectral lobes from the encoded signal and providing the remaining one of said spectral lobes as said encoded signal.

16. An encoder as claimed in claim 11, **characterized in that** the combining means comprises multiplexing means (60) for time division multiplexing the interpolated subband signals to form said encoded signal.

17. An encoder as claimed in claim 15, **characterized by** means (60) for time-division multiplexing the subband signals and the interpolation means upsamples and interpolates the multiplexed subband signals to form the interpolated subband signals.

18. An encoder as claimed in claim 15, 16 or 17, **characterized in that** the interpolation means upsamples at a rate (P) dependent upon the number of subbands (N) created by the analysis filter bank means.

19. An encoder as claimed in claim 15, 16 or 17, **characterized in that** the means (60) for time division multiplexing the subband signals comprises delay means (DB_0, DB_1, ...) for storing a series of values of each subband signal, the delay means being operable alternately between a first state wherein the delay means accepts values of the subband signals in parallel and a second state wherein the delay means outputs previously stored values serially.

A cont.

20. An encoder as claimed in claim 15, 16 or 17, **characterized in that** the means (60) for time division multiplexing the subband signals comprises first delay means (60A) and second delay means (60B) each for accepting a series of values of each subband signal, each delay means being operable alternately between a first state wherein the delay means accepts values of the subband signals in parallel and a second state wherein the delay means outputs previously stored values serially, the arrangement being such that, when the first delay means is in its first state accepting values of the subband signals, the second delay means is in its second state and outputting the subband signal values previously stored therein.

21. A decoder for decoding digital signals from an encoder as claimed in claim 11, comprising:

(iii) means (31) for extracting from the same frequency band said plurality of interpolated subband signals in the received encoded signal; and

(iv) synthesis filter means (33) complementary and substantially inverse to the analysis filter means used in encoding the received signal for processing the extracted pair of interpolated subband signals to produce a decoded signal (I′) corresponding to the input signal (I).

22. A decoder as claimed in claim 21, for decoding an encoded signal encoded by the encoder of claim 12 and further **characterized by** demodulation means (81) for orthogonally demodulating the received signal to extract the subband signals from the received signal.

23. A decoder as claimed in claim 22, for decoding an encoded signal encoded by the encoder of claim 13, and **characterized in that** the demodulation means (81) comprises means for quadrature amplitude demodulating the received encoded signal using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

C1 cont.

24. A decoder as claimed in claim 22, for decoding an encoded signal encoded by the encoder of claim 14, and **characterized in that** the synthesis filter bank (33) is arranged to compensate for the unused subband signals by substituting zero level signals.

25. A decoder as claimed in claim 21, **characterized in that** the extracting means comprises demultiplexing means (71) for time-division demultiplexing the downsampled signal to provide a plurality of received interpolated signals corresponding to said interpolated signals in the encoder.

26. A decoder as claimed in claim 25, **characterized in that** the decimator (70) downsamples the received signal and the decoder further comprises demultiplexing means (71) for time-division demultiplexing the downsampled received signal to extract the subband signals.

27. A decoder as claimed in claim 25 or 26, **characterized in that** the decoder input means downsamples at a rate (P) corresponding to the upsampling rate (P) used by the encoder to upsample the subband signals.

28. A decoder as claimed in claim 25, 26 or 27, **characterized in that** the means (71) for demultiplexing the received signal to extract the subband signals comprises delay means (DB_0, DB_1, ...) for storing a series of values of said received signal, the delay means being operable alternately between a first state wherein the delay means accepts

values of the received signal serially and a second state wherein the delay means outputs previously stored values in parallel as said subband signals.

29. A decoder as claimed in claim 25, 26 or 27, **characterized in that** the means (71) for time division demultiplexing the subband signals comprises first delay means (DB'_0) and second delay means (DB'_1) each for accepting a series of values of the received signal, each delay means being operable alternately between a first state wherein the delay means accepts values of the received signal serially and a second state wherein the delay means outputs previously stored values in parallel, the arrangement being such that, when the first delay means is in its first state accepting values of the received signal, the second delay means is in its second state and outputting the subband signal values previously stored therein.

a2 cont.

30. A method of encoding a digital input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, the encoding including the steps of:

(i) using an analysis filter bank means (21) for analyzing the input signal (I) into a plurality of subband signals ($S_0 \ldots S_{N-1}$), each subband centered at a respective one of a corresponding plurality of frequencies;

(ii) upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals each occupying the same frequency band as the others; and

(iii) combining the interpolated subband signals to form the encoded signal (E_0) for transmission or storage;

the decoding comprising the steps of:

(iv) extracting the interpolated subband signals from the received encoded signal;

(v) decimating each of the plurality of extracted interpolated subband signals to remove an equivalent number of samples to the interpolated values; and

(vi) using synthesis filter bank means (33) complementary to said analysis filter bank means (21), processing the plurality of decimated subband signals to reconstruct said input signal, **characterized in that,** during encoding, the plurality of interpolated subband signals are so combined within the encoded signal that said interpolated subband signals occupy the same frequency band.

31. A method as claimed in claim 30, **characterized in that** the combining step uses a pair of interpolated signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase-displaced by 90 degrees one relative to the other, and combines the modulated signals to provide said encoded signal (E_0), said encoded signal having two spectral lobes each comprising information from both of the interpolated signals, and the decoder extracts the modulated signals from the received encoded signal and demodulates the extracted modulated signals to produce the received interpolated signals.

32. A method as claimed in claim 31, **characterized in that** the modulation comprises quadrature amplitude modulation (QAM) using each of the interpolated subband signals to modulate a respective one of an in-phase carrier signal (f_I) and a quadrature carrier signal (f_Q), the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other, and the demodulation step at the decoder comprises the step of quadrature amplitude demodulating the received encoded signal using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

Cont

33. A method as claimed in claim 30, **characterized in that** a plurality of pairs of subband signals (S_0,S_1,S_2,S_3) are generated but only a selection (S_0,S_1) of said pairs are modulated, and the processing by the synthesis filter means compensates for the unused subband signals by substituting zero level signals.

34. A method as claimed in claim 31 or 32, further **characterized by** the step of removing one of said spectral lobes from the encoded signal.

35. A method as claimed in claim 30, **characterized in that** the interpolated signals are combined using time-division multiplexing and the extracting step time-division demultiplexes the received signal to extract the received interpolated signals.

36. A method as claimed in claim 35, **characterized in that** the step of upsampling during encoding is applied to the multiplexed interpolated signals and the step of downsampling during decoding is applied to the received signal before demultiplexing.

37. A method as claimed in claim 35 or 36, **characterized in that** the upsampling during encoding is at a rate corresponding to the number of subbands created by the analysis filter bank means.

38. A method as claimed in claim 35, 36 or 37, **characterized in that** the time division multiplexing of the interpolated signals comprises the steps of alternately storing a series of values of each subband signal, in parallel, in delay means (DB_0, DB_1, ...) and outputting serially from the delay means values previously stored therein, and the demultiplexing of the received signal to extract the subband signals comprises the steps of alternately storing a series of values of said received signal serially in delay means (DB'_0, DB'_1, ...) and outputting previously stored values in parallel as said subband signals.

39. A method as claimed in claim 35, 36 or 37, **characterized in that** the step of multiplexing the subband signals uses first delay means (60a) and second delay means (60b) each for accepting a series of values of each subband signal, each delay means being alternately loaded with values of the subband signals in parallel and outputting previously stored values serially, the arrangement being such that, during the step of loading each of the delay means with values of the subband signals, the other delay means is outputting the subband signal values previously stored therein, and the step of time division demultiplexing of the subband signals uses first delay means (71A) and second delay means (71B) each for accepting a series of values of the received signal, each delay means being alternately loaded with values of the received signal serially and outputting previously stored values in parallel, the arrangement being such that, when each of the delay means is being loaded with values of the received signal, the other of the delay means is outputting the subband signal values previously stored therein.

~~40.~~ A method of encoding a digital input signal for transmission or storage comprising the steps of:

Cont.

(i) using an analysis filter bank means (21), analyzing the input signal into a plurality of subband signals, each subband centered at a respective one of a corresponding plurality of frequencies;

(ii) upsampling and interpolating each subband signal to provide a plurality of interpolated signals each occupying the same frequency band as the others; and

(iii) combining the interpolated subband signals to form the encoded signal for transmission or storage, **characterized in that** the plurality of subband signals are so combined within the encoded signal that said interpolated subband signals occupy the same frequency band.

G1 cont.

41. An encoding method as claimed in claim 40, **characterized in that** the combining step uses said pair of interpolated subband signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase displaced by 90 degrees one relative to the other, and combines the modulated signals to provide said encoded signal, such that the encoded signal has two spectral lobes, each comprising information from both of the interpolated subband signals.

42. An encoding method as claimed in claim 41, **characterized in that** the combining step uses each of the interpolated subband signals for quadrature amplitude modulation of a respective one of an in-phase carrier signal (f_I) and a quadrature carrier signal (f_Q), the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other.

43. An encoding method as claimed in claim 40, **characterized in that** a plurality of pairs of subband signals (S_0,S_1,S_2,S_3) are generated using the analysis filter means but only a selection (S_0,S_1) of said pairs are modulated.

44. An encoding method as claimed in claim 41, 42 or 43, further **characterized by** the step of removing one of said spectral lobes from the encoded signal.

45. An encoding method as claimed in claim 40, **characterized in that** the interpolated subband signals are combined using time-division multiplexing.

46. An encoding method as claimed in claim 45, **characterized in that** the upsampling is performed after the step of multiplexing the subband signals.

47. An encoding method as claimed in claim 45 or 46, **characterized in that** the upsampling is at a rate (P) corresponding to the number of subbands (N) provided by the analysis filter bank means.

48. An encoding method as claimed in claim 45, 46 or 47, **characterized in that** the multiplexing of the subband signals uses delay means (DB_0, DB_1, ...) for storing a series of values of each subband signal, the values being stored in the delay means in parallel and, alternately, previously stored values outputted from the delay means serially.

49. An encoding method as claimed in claim 45, 46 or 47, **characterized in that** the step of multiplexing the subband signals uses first delay means (60A) and second delay means (60B) each for accepting a series of values of each subband signal, each delay means being loaded with values of the subband signals in parallel and, alternately, outputting previously stored values serially, the arrangement being such that, during loading of each of the delay means with values of the subband signals, the other delay means is outputting the subband signal values previously stored therein.

50. A method of decoding an encoded signal encoded by the encoder method of claim 40, comprising the steps of:

(iii) extracting said pair of interpolated signals from the same frequency band and from a received encoded signal (E'_0); and

(iv) using synthesis filter bank means (33) complementary and substantially inverse to the analysis filter bank means used during encoding, to process the extracted pair of received interpolated signals to produce a decoded signal (I') corresponding to the input signal (I).

Cont.

51. A decoding method as claimed in claim 50, for decoding an encoded signal encoded by the encoding method of claim 41 and further **characterized by** the step of orthogonally demodulating the received signal to extract the interpolated signals.

52. A decoding method as claimed in claim 51, for decoding an encoded signal encoded by the encoding method of claim 42, **characterized in that** the demodulating of the received encoded signal comprises quadrature amplitude demodulation using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

53. A decoding method as claimed in claim 50, for decoding an encoded signal encoded by the encoder of claim 43, and **characterized in that** the processing using the synthesis filter bank means is arranged to compensate for unused subband signals by substituting zero level signals.

54. A method as claimed in claim 50, **characterized in that** the interpolated signals are combined using time-division multiplexing.

55. A decoding method as claimed in claim 54, **characterized in that** the received encoded signal is downsampled before demultiplexing.

56. A decoding method as claimed in claim 54 or 55, **characterized in that** the downsampling is at a rate (P) corresponding to the upsampling rate (P) used during encoding.

57. A decoding method as claimed in claim 54, 55 or 56, **characterized in that** the step of demultiplexing the received signal to extract the interpolated signals comprises the steps of alternately storing a series of values of said received signal serially in delay means (DB'_0, DB'_1 ...) and outputting previously stored values in parallel as said interpolated signals.

58. A decoding method as claimed in claim 54, 55 or 56, **characterized in that** the step of demultiplexing the interpolated signals uses first delay means (71A) and second

Cont.

a' Cont.

delay means (71B) each for accepting a series of values of the received signal, each delay means having values of the received signal loaded therein serially and, alternately, outputting previously stored values in parallel, the arrangement being such that, when the values of the received signal are being loaded into one of the delay means, the interpolated signal values previously stored in the other delay means are being outputted.

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PATENT COOPERATION TREATY

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INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

Applicant's or agent's file reference AP582PCT	**FOR FURTHER ACTION**	See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)
International application No. PCT/CA98/00816	International filing date *(day/month/year)* 28/08/1998	Priority date *(day/month/year)* 29/08/1997
International Patent Classification (IPC) or national classification and IPC H04L27/00		
Applicant BELL CANADA et al.		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 8 sheets, including this cover sheet.

   ☐ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of sheets.

3. This report contains indications relating to the following items:

   I ☒ Basis of the report
   II ☐ Priority
   III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   IV ☐ Lack of unity of invention
   V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations suporting such statement
   VI ☐ Certain documents cited
   VII ☒ Certain defects in the international application
   VIII ☒ Certain observations on the international application

Date of submission of the demand 15/03/1999	Date of completion of this report 08.07.99
Name and mailing address of the international preliminary examining authority: European Patent Office D-80298 Munich Tel. (+49-89) 2399-0 Tx: 523656 epmu d Fax: (+49-89) 2399-4465	Authorized officer Keller, M Telephone No. (+49-89) 2399

I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

Description, pages:

1-23 as originally filed

Claims, No.:

1-58 as originally filed

Drawings, sheets:

1/21-21/21 as originally filed

2. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:
- ☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. Statement

Novelty (N)	Yes:	Claims	1-58
	No:	Claims	
Inventive step (IS)	Yes:	Claims	1-58
	No:	Claims	
Industrial applicability (IA)	Yes:	Claims	1-58
	No:	Claims	

2. Citations and explanations

see separate sheet

VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

see separate sheet

VIII. Certain observations on the international application

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

see separate sheet

With respect to SECTION V:

The present international application PCT/CA98/00816 relates, according to the title established by the International Searching Authority, to digital transmission using subband coding.

Claim 1 claims an apparatus comprising an encoder and a decoder,
independent Claim 11 claims the encoder,
independent Claim 21 claims the decoder,
independent Claim 30 claims a method of encoding a digital input signal and decoding such encoded signal,
independent Claim 40 claims the method of encoding, and
independent Claim 50 claims the method of decoding.

The nearest **prior art** according to the International search report is represented by the document
D1 = SANDBERG, TZANNES: "Overlapped discrete multitone modulation for high speed copper wire communications" IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, vol. 13, no. 9, December 1995, pages 1571-1585, XP000543156 New York, US
which is already acknowledged in the international application, on page 2.

The problems to be solved and the advantages achieved by the present international application are mainly explained on page 1 (line 14) to page 5 (line 2) and on page 19, lines 15 to 33. In particular, the present international application seeks to mitigate the disadvantages of the known systems which are susceptibility to distortion and corruption by impulse noise.

The problems are solved and the advantages are achieved
with respect to Claim 1
by an **apparatus** comprising
an encoder (12)
for encoding a digital input signal for transmission or storage and
a decoder (17)
for decoding such encoded signal to reconstruct the input signal,

the encoder comprising

- analysis filter bank means (21)
  for analyzing the input signal into a plurality of subband signals,
  each subband centered at a respective one of a corresponding plurality of frequencies,
- interpolation means (22; 62)
  for upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals all occupying the same frequency band; and
- combining means (23; 60; 74)
  for combining the interpolated subband signals to form the encoded signal for transmission or storage;

and

the decoder comprising

- synthesis filter bank means (33)
  complementary to said analysis filter bank means for producing a decoded signal corresponding to a reconstruction of the input signal,
- extraction means (31; 71; 81)
  for extracting the interpolated subband signals from the received encoded signal; and
- decimator means (32; 70)
  for decimating each of the plurality of extracted interpolated subband signals to remove a number of samples corresponding to those interpolated during encoding and applying the decimated signals to the synthesis filter bank means (33), the synthesis filter bank means processing the plurality of decimated signals to reconstruct said input signal,

characterized in that,

in the encoder, the combining means (23; 60; 74) so combines the plurality of interpolated subband signals within the encoded signal that said interpolated subband signals occupy the same frequency band and

the decoder separates the interpolated subband signals from within said same frequency band.

Independent Claims 11 and 21 seek individual protection for the encoder and the decoder respectively according to the specification made in the apparatus of Claim 1.

The problems are solved and the advantages are achieved with respect to independent Claim 30 by a **method of encoding a digital input signal** for transmission or storage **and decoding** such encoded signal to reconstruct the input signal,

the *encoding* including the steps of:

(i) using an analysis filter bank means (21) for analyzing the input signal (I) into a plurality of subband signals (S_0 ... S_{N-1}),
each subband centered at a respective one of a corresponding plurality of frequencies;

(ii) upsampling and interpolating each subband signal to provide a plurality of interpolated subband signals each occupying the same frequency band as the others; and

(iii) combining the interpolated subband signals to form the encoded signal (Eo) for transmission or storage;

the *decoding* comprising the steps of:

(iv) extracting the interpolated subband signals from the received encoded signal;

(v) decimating each of the plurality of extracted interpolated subband signals to remove an equivalent number of samples to the interpolated values; and

(vi) using synthesis filter bank means (33) complementary to said analysis filter bank means (21), processing the plurality of decimated subband signals to reconstruct said input signal,

characterized in that,
during encoding, the plurality of interpolated subband signals are so combined within the encoded signal that said interpolated subband signals occupy the same frequency band.

Independent Claims 40 and 50 seek individual protection for the encoding method and the decoding method respectively which are comprised in the method of Claim 30.

Neither the above-mentioned document nor the additionally cited documents of the search report disclose or suggest such features. Thus, the subject-matter

defined in the above-mentioned independent claims is considered to be novel and to involve an inventive step (Articles 33 (2) and (3) PCT).

Dependent Claims 2 to 10, 12 to 20, 22 to 29, 31 to 39, 41 to 49 and 51 to 58 relate to further details of the devices and methods respectively and are therefore equally novel and inventive (Art. 33 (2) and (3) PCT).

Industrial applicability (Article 33 (4) PCT) of the subject-matter claimed is beyond doubt. Moreover, industrial applicability is explained on page 23 of the description.

With respect to SECTION VII:

The following remarks are of minor relevance and should be taken into consideration when entering the national or regional phase.

1. The applicant has acknowledged articles on page 3, line 5 by the term "**incorporated herein by reference**". When entering the national or regional phase, either the phrase "incorporated herein by reference" should be deleted or the cited relevant prior art under Rule 5.1 (a)(ii) PCT could be discussed in a short summary (see furthermore PCT INTERNATIONAL PRELIMINARY EXAMINATION GUIDELINES, 29.10.1998, S-07/1998 (E), Section IV, II-4.17).

2. On page 21, line 25, reference is made to the "spirit of the present invention". This statement is obviously unnecessary and therefore should be excised from the application (Rule 9.1 (iv) PCT; see also PCT Guidelines chapter III, 4.3a).

With respect to SECTION VIII:

The following remarks are of minor importance and should be taken into consideration when entering the national or regional phase.

1. Dependent Claim 9 refers to "An <u>encoder</u> as claimed in claim 6, 7 or 8, ...". However, Claim 1 as well as dependent Claims 6 to 8 relate to an *apparatus*. In order to keep the terminology consistent, it is proposed to amend the wording in that dependent Claim 9 might read as follows: "Apparatus as claimed in claim 6 or 7, characterized in that in the encoder, the means (60) for time division multiplexing the subband signals comprises ...". Moreover it should be noted that the reference to dependent Claim 8 in present Claim 9 is irritating because present Claim 8 does neither refer to nor include an encoder.

2. For the sake of a consistent terminology (cf. Rule 10.2 PCT), independent Claim 11 should read in the characterizing part as follows: "the combining means (23; 60; 74) so combines the plurality of <u>*interpolated*</u> subband signals within the encoded signal that said interpolated subband signals occupy the same frequency band."

3. Independent Claim 21 should refer to
   (i) means (31) for extracting from the same frequency band ...
   (ii) synthesis filter **bank [** means **]** (33) ...
   for the sake of a consistent terminology (Rule 10.2 PCT) and a consecutive indexing (Art. 6 PCT).

PATENT COOPERATION TREATY

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INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

Applicant's or agent's file reference AP582PCT	**FOR FURTHER ACTION** see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below.	
International application No. PCT/CA 98/ 00816	International filing date *(day/month/year)* 28/08/1998	(Earliest) Priority Date *(day/month/year)* 29/08/1997
Applicant BELL CANADA et al.		

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 3 sheets.

[X] It is also accompanied by a copy of each prior art document cited in this report.

1. [ ] **Certain claims were found unsearchable** (see Box I).

2. [ ] **Unity of invention is lacking** (see Box II).

3. [ ] The international application contains disclosure of a **nucleotide and/or amino acid sequence listing** and the international search was carried out on the basis of the sequence listing
   - [ ] filed with the international application.
   - [ ] furnished by the applicant separately from the international application,
     - [ ] but not accompanied by a statement to the effect that it did not include matter going beyond the disclosure in the international application as filed.
   - [ ] Transcribed by this Authority

4. With regard to the **title**,
   - [ ] the text is approved as submitted by the applicant
   - [X] the text has been established by this Authority to read as follows:

   DIGITAL TRANSMISSION USING SUBBAND CODING

5. With regard to the **abstract**,
   - [X] the text is approved as submitted by the applicant
   - [ ] the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this International Search Report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is:

   Figure No. 2
   - [X] as suggested by the applicant.
   - [ ] because the applicant failed to suggest a figure.
   - [ ] because this figure better characterizes the invention.
   - [ ] None of the figures.

INTERNATIONAL SEARCH REPORT

International Application No
PCT/CA 98/00816

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04L27/00 H04B1/66

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04L H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	SANDBERG,-TZANNES: "Overlapped discrete multitone modulation for high speed copper wire communications" IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS, vol. 13, no. 9, December 1995, pages 1571-1585, XP000543156 New York, US cited in the application see page 1573, left-hand column - right-hand column, paragraph 2 see page 1574, left-hand column, paragraph 1 --- -/--	1,11,21, 30,40,50

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
29 January 1999	05/02/1999

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Scriven, P

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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
A	GANDHI ET AL.: "Wavelets for baseband coding of waveforms" GLOBAL TELECOMMUNICATIONS CONFERENCE, 28 November 1994 - 2 December 1994, pages 363-367, XP000488574 New York, US see page 364, right-hand column, paragraph 2 - paragraph 4 see page 365, left-hand column, paragraph 4 see page 365, right-hand column, paragraph 1 - paragraph 3 ---	1,11,21, 30,40,50
A	GROSS ET AL.: "Discrete wavelet multitone (DWMT) system for digital transmission over HFC linkes" PROCEEDINGS OF THE SPIE, vol. 2609, 23 October 1995, pages 168-175, XP000576748 Bellingham, US see page 169, paragraph 2 - paragraph 3 ---	1,11,21, 30,40,50
A	YANG ET AL.: "A multirate wireless transmission system using wavelet packet modulation" IEEE VEHICULAR TECHNOLOGY CONFERENCE, 4 - 7 May 1997, pages 368-372, XP000701822 New York, US see page 370, left-hand column, paragraph 3 ---	1,11,21, 30,40,50
A,P	WO 98 09383 A (BELL CANADA) 5 March 1998 cited in the application see abstract ---	1-58
A,P	AKANSU ET AL.: "Wavelet and subband transforms: fundamentals and communication applications" IEEE COMMUNICATIONS MAGAZINE, vol. 35, no. 12, December 1997, pages 104-115, XP000770959 New York, US cited in the application see figure 11 -----	1,11,21, 30,40,50

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/CA 98/00816

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9809383 A	05-03-1998	CA 2184541 A	01-03-1998
		AU 4007297 A	19-03-1998